Powersoft

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POWERSOFT KDSP
USER MANUAL

EDITION Sept. 2007

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REVISIONS

REVISION	DATE	REVISION PURPOSE	Signed by
1.2	01/2007	1st Edition	T Morganti
1.3	03/2007	Sticky preset names	T Morganti
1.4	04/2007	Amp name, Menu locking, Reorgan.	T Morganti
1.4.3	09/2007	Removed general amp news. Removed some “todo” parts (PCM, filters definition, howtos)	T Morganti

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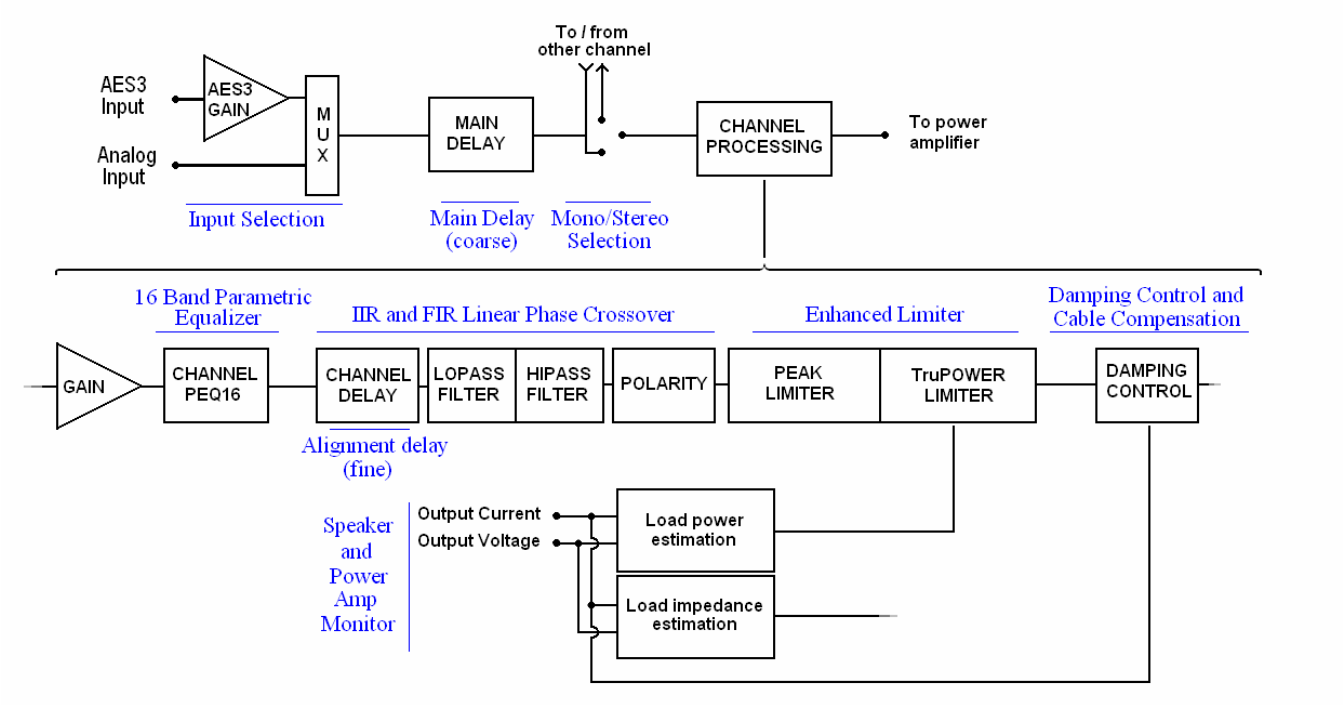
1. DSP PROCESSING CHAIN

The KDSP board embeds a very advanced sound processor. There are some features found only in high level sound processor, and some new functionalities that can be implemented only if a complete DSP/amplifier integration is done, as for the Powersoft KSeries amplifier.
A complete block diagram of the processing chain is shown below.



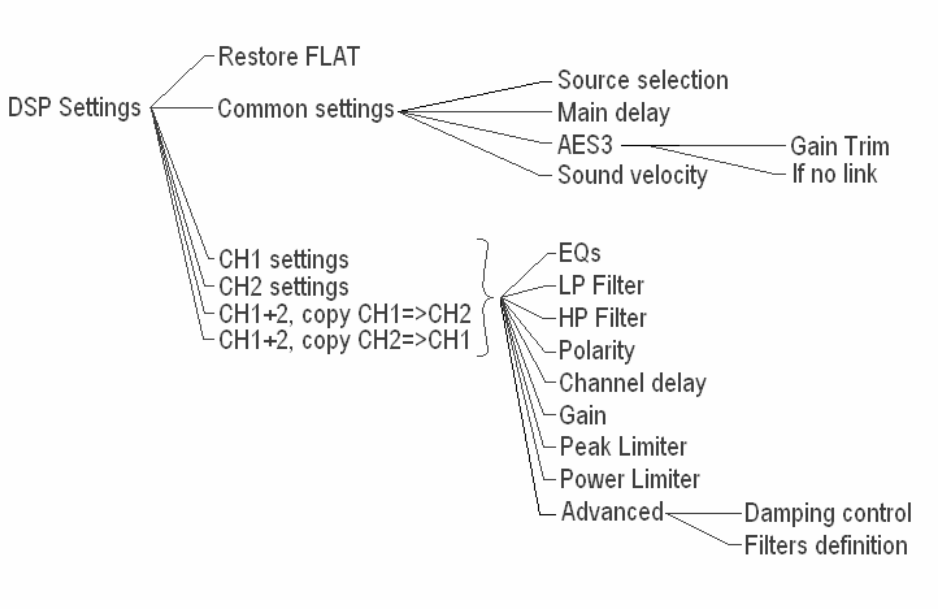
The features of the current DSP release are:

- ✓ AES3 digital input
- ✓ high performance (128 dB SNR) analog input: get the optimal quality whatever is your level
- ✓ high performance (122 dB SNR) analog output: no quality loss with DSP inserted in the path
- ✓ latest generation SHARC DSP with 40 bit floating point signal path for noiseless processing
- ✓ enhanced IIR filtering routine optimized for the highest SNR
- ✓ up to 2 seconds of main delay
- ✓ PEQ16: fully parametric equalizer, 16 band, each with 8 different filter shapes (peaking, hi/lo shelving, hi/lo pass EQ, band pass, band stop, all pass)
- ✓ alignment delay with high temporal resolution
- ✓ crossover with IIR Butterworth, Bessel and Linkwitz-Riley filters
- ✓ advanced crossover with FIR linear phase brickwall filters
- ✓ speaker and amplifier monitor and analizer to estimate power and load impedance
- ✓ peak limiter with lookahead
- ✓ NEW: true power limiter to avoid burning voice coils while getting the maximum performances from drivers
- ✓ NEW: **patented** damping control and cable resistance compensation: get optimally damped bass response with any cabling length

and more coming next in future versions…

2. KDSP SETTINGS

KDSP SUBMENU



RESTORE FLAT

This command resets to default values all the DSP parameters, after a confirmation request. Flat frequency response, 0 dB gain, + polarity, no delay, no limiters, no damping control.
The parameters belonging to input signal selection, such as the stereo/parallel input mode and the AES3 gain trim are NOT modified.

COMMON SETTINGS

These parameters are applied (identical) to both channels.

SOURCE SELECTION

This menu allows to choose the input selection mode.
In “Stereo Mode” the signal on CH1 IN is processed and then goes to CH1 OUT, and that on CH2 IN is processed and then goes to CH2 OUT.
In “Parallel from CH1” the signal on CH1 IN goes on both CH1 and CH2 out, through indipendent DSP processing. In similar way, in “Parallel from CH2” mode the source is the signal on CH2 IN.

MAIN DELAY

This is the delay that is applied to both channels. This is mainly intended for delayed towers. The delay ranges from 0 to 2000 ms, with 1 ms stepping.

AES3

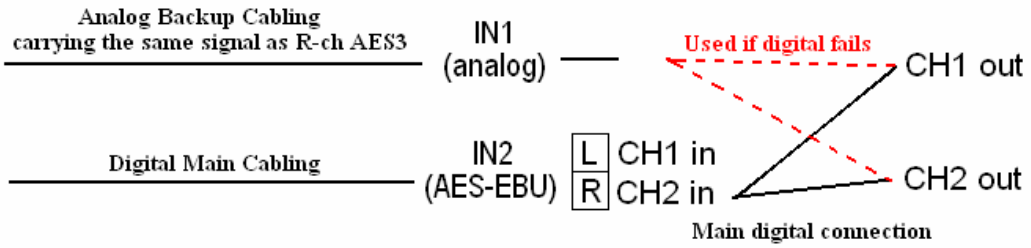
This menu controls the AES3 input options.

GAIN TRIM

This menu sets the gain to be applied at the signal coming from the AES3 digital input. Setting 0 dB makes a full-scale digital signal equivalent to an analog input signal of 20 dBu.

IF NO LINK

This menu controls the behaviour if the AES3 signal connection fails or becomes unreliable (>1% of errors). You can choose either "mute" or "analog".
If "mute" is selected, when the AES3 connection fails, the amplifier will mute the output.
If "analog" is selected, when the AES3 connection fails, the amplifier will take the signal from the analog input. The switching is done in real time in order to avoid any glitch. If the input levels are correctly matched between analog input and AES3 input (use AES3 GAIN TRIM parameter), the switching between AES3 and analog will be inaudible.
When you use the analog input backup, you must run the amplifier in parallel mode to select only one digital input channel and you have to connect the corresponding analog signal cable in the ANALOG IN1 plug. There is only one working way to cable the amplifier:



Analog backup mode connection: the amplifier runs with AES3 input selected, in parallel mode. In the above example diagram, the R channel of the AES3 stream is selected, which is the "Parallel from CH2" choice from Source Selection submenu. Also the L channel can be used, by selecting "Parallel from CH1". This signal goes, through processing, to both amplifier outputs. A cable with an analog signal identical to that provided by AES3 should be connected to IN1 of the amplifier for backup purposes. The level should be accurately trimmed, by using the AES3 gain trim setting, or by changing the analog signal level.

SOUND VELOCITY

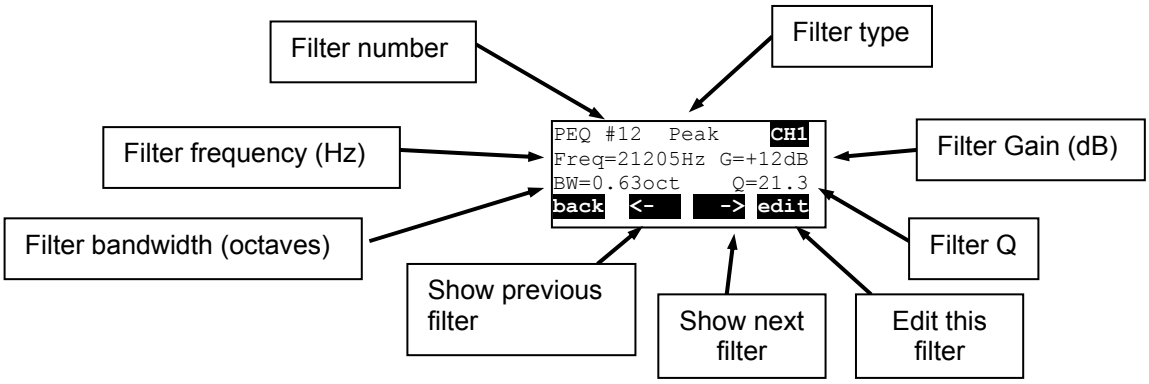
This option sets the sound velocity used for time to distance conversions troughout the local interface. It can be set from 320 m/s to 360 m/s.

CH1, CH2, CH1+2 SETTINGS

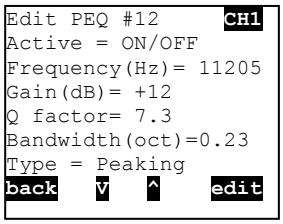
In all these submenus, the channel (or the channels) that are being edited are shown on the top-right corner: CH1, CH2, or 1+2 when you are editing both channels at once.

EQs

This menu enters the parametric equalizer input interface, that lists the 16 parametric filters one by one. The current filter number is shown on the left of the first line. By pressing the arrows, you can move from one filter to the next or to the previous. The parameters are sintetized on the display: you can check the settings of the whole equalizer by simply move through all the 16 filters.



By clicking "edit", you can change the settings for the selected filter.



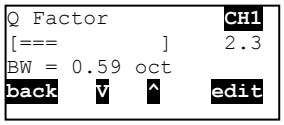
You can enable or disable (flat response) the filter.
You can choose the filter type from the following types:

- ✓ peaking (Peak)
- ✓ low (LSh) and high (HSh) shelving (from 3 to 15 dB/oct)
- ✓ low pass EQ (LPEQ) and high pass EQ (HPEQ)
- ✓ bandstop (Notch)
- ✓ bandpass (BandP)
- ✓ allpass (AllP)

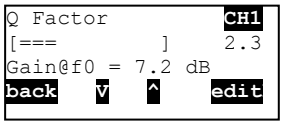
You can change the frequency (20 to 20000Hz, 1/96 octave step), the gain (only if filter is peaking or shelving, from –15 to +15 dB, 0.1 dB step), the quality factor (for all filters but shelving, 0.1 to 30, 0.1 step), and the slope (only for shelving, from 3 to 15 dB/oct).

For the exact definition of these parameters and for having a look to the frequency response of the filters, please read the last part of this manual.

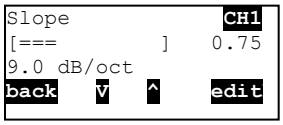
Peaking/Bandpass/Bandstop/Allpass



Lowpass/Highpass EQ



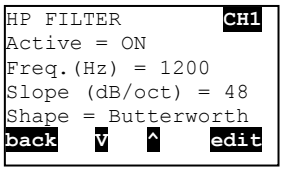
Shelving

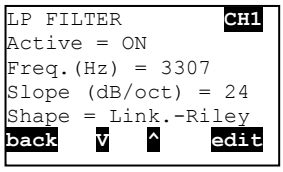


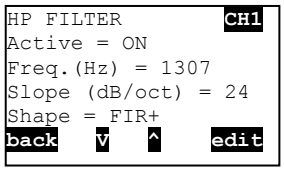
Q/Bandwidth, Q/Gain and shelving slope dual mode input.

LP FILTER and HP FILTER

This menu sets up the crossover filters. There are 2 available filters, one lowpass filter and one highpass filter. By using both, you will obtain a bandpass response. Both traditional IIR and brickwall linear phase FIR are implemented.
The classic IIR crossover filters can be selected as a Butterworth shape, Bessel shape and Linkwitz-Riley shape. In the first 2 cases, the Freq. parameter defines the –3 dB point, in the latter, the –6 dB point. The slope is freely selectable from 6 dB/octave (1st order filter) to 48 dB/octave (8th order filter).
The FIR filters can be selected as normal (FIR) or enhanced (FIR+). The enhanced version gives a higher rejection of out of band signals, at the expense of a little (<10 degrees) phase nonlinearity. In both cases, the minimum working frequency is related to the desired latency, and with standard settings (2.0 ms) is limited to about 400 Hz. So it is better to use the FIR filters to crossover upper midranges or mid-high drivers, for which the phase coherency is a key point. The use of the FIR filters causes the deactivation of the limiter lookahead feature, in order to preserve the overall 2.0 ms latency.







POLARITY

This menu allows to reverse the signal polarity. It is equivalent to a 180° phase change.

CHANNEL DELAY

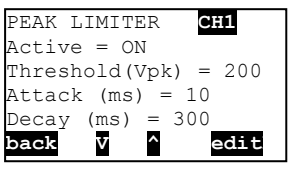
This menu allows to time-align two different transducers on the two amplifier channels. The selectable delay varies from 0 to 30 ms (about 10 meters), with one sample step (1/96000th second or 10.4 us, about 3.5 mm) step.

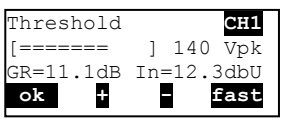
GAIN

This menu changes the channel gain, from –40 dB to +15 dB, with a 0.1 dB step.

PEAK LIMITER

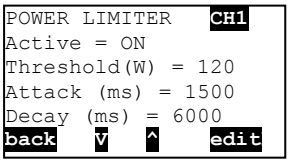
This menu controls the output peak limiter. The peak limiter decreases the gain in order to reduce the maximum output voltage. The parameters are the voltage threshold at which the gain begins to get reduced, the attack time, that is the response time of the limiter intervention and the decay time, that is the time constant after which the gain is restored to the nominal gain. You should in first setup the time parameters, and then you should adjust the threshold voltage. When editing the threshold, the display shows in real-time the gain reduction due to the limiting function (GR = xx.x dB), togheter with the limiting voltage referred to the input.
The limiter is conceived to pass-through inalterated the signal peaks lasting less than the attack time parameter, in order to exploit the exceptional dynamics of the K-Series amplifier.
The limiter has also a lookahead buffer, in order to soften the clipping and to minimize the distortion, giving a superior sonic performance. The default value of the lookahead time is 1.5 ms. When FIR filter are activated, the limiter lookahead feature is deactivated in order to preserve the overall 2.0 ms latency.

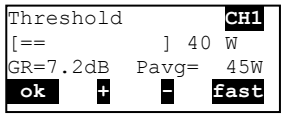




POWER LIMITER

This menu controls the output power limiter. The power limiter decreases the gain in order to reduce the power delivered to the load. The unusual ability to measure the load current gives the opportunity to estimate the actual active power output by the amplifier. Given the low efficiency of electromechanical transducers (<10%), almost the same level of power is transformed into heat into the voice coil. This limiter is intended to avoid burning the voice coils of the drivers while exploiting their maximum performances.
The parameters are the power threshold in W at which the gain begins to get reduced, the attack time, that is the response time of the limiter intervention and the decay time, that is the time constant after which the gain is restored to the nominal gain. You should select as power threshold the true power handling of your speaker (it is usually between one third and one half of the declared RMS or AES power handling, see the table below), with time constants related to the thermal inertia of your speaker (the mass to be heated). So, huge 4" VC subwoofers could require some seconds of attack and some seconds of release time, while tiny 1' VC tweeters may need less than 1/10 of second of attack time. When editing the threshold, the display shows in real-time the gain reduction due to the combined peak/power limiting function (GR = xx.x dB) and the power really delivered to the load (Pavg=xxxxW).





DRIVER	POWER THRESHOLD	ATTACK TIME	DECAY TIME
1" VC tweeter	10…20 W	100 ms	300 ms
1.5" VC tweeter	20…30 W	150 ms	300 ms
2" VC horn driver	20…40 W	200 ms	400 ms
3" VC horn driver	30…50 W	300 ms	500 ms
4" VC horn driver	40…60 W	500 ms	3000 ms
2" VC midrange	30…100 W	500 ms	3000 ms
3" VC midbass	50…150 W	1000 ms	5000 ms
4" VC woofer	100…200 W	2000 ms	5000 ms
4" VC subwoofer	150…250 W	4000 ms	8000 ms
6" VC massive subwoofer	250…500 W	6000 ms	10000 ms

Suggested power limiter settings for common used drivers.

The power limiter should be used to protect the drivers from burning, and at working levels it normally shouldn't activate. Check the gain reduction, it should not raise over 2-4 dB even in the loudest piece of music, in order to obtain the optimal sound. Note that a normal musical signal has very loud peaks, but a small average level, while a continuous tone has a much higher average power even if it has a modest impact on the hear.

ADVANCED CHANNEL SETTINGS

These are advanced settings that are not usually found on any other sound processor or amplifier.

DAMPING CONTROL

This unique and **patented** feature allows to add a "virtual" resistor in series with the amplifier output. The value of this resistor can also be negative in order to get the correct damping factor with any cabling you use. For example, if you use a 10 meters cable for powering the subs, you will add a parasitic series resistance (0.3 ohm for instance). By enabling the damping control, you can add a negative resistance that compensates the cable resistance.
WARNING: when enabled, a lowpass filter cutting around 400 Hz is automatically inserted into the chain. This feature is intended to be used only for subwoofer applications.

AWG	Sq mm	Length	Resistance
16	2 x 1.5 mm2	5 m	0.13 Ohm
16	2 x 1.5 mm2	10 m	0.26 Ohm
16	2 x 1.5 mm2	20 m	0.52 Ohm
14	2 x 2.5 mm2	5 m	0.08 Ohm
14	2 x 2.5 mm2	10 m	0.16 Ohm
14	2 x 2.5 mm2	20 m	0.32 Ohm
12	2 x 4 mm2	5 m	0.05 Ohm
12	2 x 4 mm2	10 m	0.10 Ohm
12	2 x 4mm2	20 m	0.20 Ohm

Typical cabling resistance.

Another advantage is that you can take in account the voice coil rise in resistance due to heating, getting a correctly damped bass response in the average working condition. For example, if you will run hard your subwoofer amplifiers during your performance, you should add a negative resistance between 1 and 2 Ohms to keep the correctly damped response. This value is correct for an 8 Ohm typical subwoofer, remember to divide it for the number of paralleled speakers on the K amplifier output. Take into account that if you use the system at a lower level, you should select a smaller negative resistance to avoid an overdamped response.

Average power / rated power	Power compression	Equivalent series resistance for 8 Ohm driver
10%	1.4 dB	1.0 Ohm
20%	2.0 dB	1.4 Ohm
50%	2.8 dB	2.1 Ohm
100%	4.5 dB	3.8 Ohm

Typical rise in resistance due to voice coil heating.
Note the huge amount (3.8 Ohm) when the driver reaches its thermal limit.

FILTERS DEFINITION

This feature allows to emulate the parameters input of some commercial audio processors, in order to get the same equalizer frequency response by simply inputting the same parameter values. You should not learn some complicated conversion formulas or frequency response definitions to get the same behaviour of the processor you are used to work with.
In the current version, the supported modes are Powersoft (Type1) and alternative (Type2), contact Powersoft support for correlation with other Processing Units from other manufacturers.

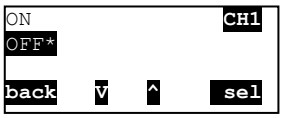
CHANNEL SETTINGS COPY

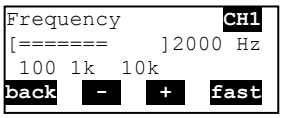
By selecting the "CH1+2, copy CH1=>CH2" submenu, all the CH2 DSP settings are cloned from the CH1, and then the parameters of both channels are changed at the same time.
The same for "CH1+2, copy CH2=>CH1".

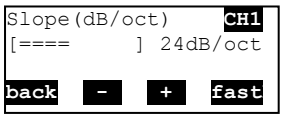
3. *Howto*

Setup a crossover

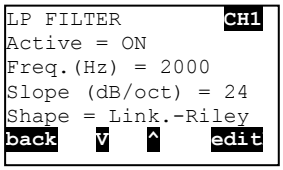
For example, midrange on CH1 and high frequency driver on CH2.
Edit CH1, go to LP Filter, set Active=ON, set Frequency = 2000 Hz, set Shape=Linkwitz-Riley, set Slope=24dB/oct. You could also experiment with shape=FIR.



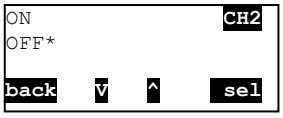


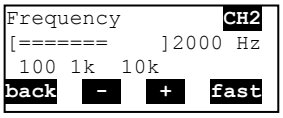


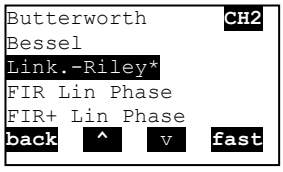
Final result:

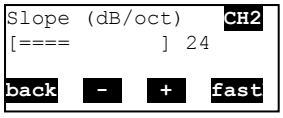


Edit CH2, go to HP Filter, set Active=ON, set Frequency = 2000 Hz, set Shape=Linkwitz-Riley (or FIR), set Slope=24dB/oct. You could also experiment with shape=FIR.

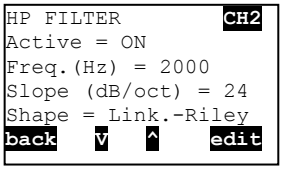








Final result:



On CH2, edit Ch Delay to temporal align midrange with highs horn.
On CH2, edit Gain to align levels.

Setup AES3 input with analog backup

Refer to IF NO LINK command.

Setup a bridged configuration

If you already have a preset made for a non-bridged configuration, lets say for the CH2 for example, you should load the preset, then choose to edit “CH1+2, copy CH2=>CH1”. The CH1 now is identical to CH2. With this command you can also edit both channels at the same time.
Otherwise, you can right away choose “CH1+2, copy CH1=>CH2” or “CH1+2, copy CH2=>CH1”, then edit both channels at once.
Remember that, in bridged mode, the 2 DSP channels should be identical to get the right results.

4. *Addendum*

Eq Filters magnitude responses